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EXAMINER	
MEI, XU	
ART UNIT	**PAPER NUMBER**
2615	

DATE MAILED: 10/06/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/473,547	BENESTY ET AL.
	Examiner	Art Unit
	Xu Mei	2615

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 7/25/2006.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-52* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-3,6-8,11-13,18-20,25-28,33-35,38-40 and 46-48* is/are rejected.

7)☒ Claim(s) *4-5, 9-10, 14-17, 21-24, 29-31, 36-37, 41-45, and 49-52* is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

1. This communication is responsive to the applicant's amendment dated 07/25/2006.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims **1, 6, 11, 18, 26, and 33** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. (US 5,396,554, hereafter, Hirano) in view of Nam et al (IEEE Paper of 1990, hereafter, Nam).

Regarding **claim 11,** Hirano disclose in Fig. 3 a multi-channel echo canceling apparatus (100) for transmitting a signal over a channel (26) in a multiple-channel communication apparatus where said signal includes an input signal (22) and multiple "impulse responses" (i.e., echoes 15 and 16) (The acoustic echo signals reproduced by loudspeakers 13 and 14,

convolved with the impulse responses of acoustic echo paths 15 and 16, respectively, and then received by microphone 19 may be loosely described as "containing the impulse responses" of the acoustic echo paths, the apparent meaning intended by Applicant.), wherein said multiple impulse responses (echoes) are to be adaptively filtered (column 9, lines 24-27), said apparatus comprising:

a transmitter (inherently) for generating a data signal for transmission via a communication channel (26), wherein said signal includes an input signal (22) and multiple impulse responses (acoustic echo signals 15 and 16) wherein said multiple impulse responses (echoes) are to be adaptively filtered (column 9, lines 24-27);

an adaptive filter circuit (103) for generating an estimate of an impulse response corresponding to each of said [multiple] impulse responses (column 9, lines 50-54);

a subtracter circuit (105) for generating an error signal (output signal 26) representing the difference between said data signal (24) and a sum of said estimates (adaptive filter 103 simultaneously generates a combined echo ["impulse response"] estimate signal that is a sum of estimates of the echo signals ["impulse responses"] contained in data signal 24 due to acoustic echo paths 15 and 16, based on the assumption that one

of the echo signals is simply a delayed replica of the other, as described at column 4, lines 1-19);

wherein said estimates are generated using a time domain recursive least squares algorithm (Hirano discloses at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter may operate in the frequency domain), but not specifically shows is the RLS algorithm is a frequency domain RLS algorithm.

Nam discloses a frequency domain block RLS adaptive algorithm that is useful for identification of nonlinear system that using adaptive filtering process and for nonlinear distortion in communication system (i.e., echo system). And the frequency domain block RLS adaptive algorithm has the advantage as a power technique in transfer function approach for obtaining statistically meaningful results using the frequency domain block RLS adaptive algorithm (Section 4, Conclusion on page 2409-2410 of the IEEE Paper).

Therefore, it would have been obvious to one of ordinary skill in the art to modify the multi-channel echo canceling apparatus of Hirano with a more powerful frequency domain block RLS adaptive algorithm as taught by Nam in adaptive filtering transfer function processing in order to obtaining more accurate statistically meaningful results.

Regarding **claims 1 and 6**, in normal operation, the apparatus of Fig. 3 of Hirano clearly performs the methods claimed, according to the description of Hirano and Nam as discussed above regarding claim 11.

Regarding **claims 18, 26, and 33**, Hirano disclose in Fig. 2 a prior-art system and associated inherent method of multi-channel communication (comprising canceling acoustic echo distortion in a communication system) between at least first and second locations (as generally disclosed, e.g., column 1, lines 13-50), said method comprising the steps of:

transmitting multiple channels of information (501 and 502) upstream from said first location (not illustrated, but inherently present, providing received signals 501 and 502 and receiving transmission signals 516 and 517) to said second location (as illustrated);

transmitting at least one additional channel (516) of information downstream from said second location to said first location;

generating estimates (the impulse responses of adaptive filters 531 and 532) of impulse responses (developing an estimated impulse response) corresponding to distortion paths (corresponding to each of said [channels from the first location

to the second location] 501 and 502, and that models an interference path at said second location from said corresponding [channel from the first location to the second location] to said [channel from said second location to said first location]) (acoustic echo paths 505 and 506) at said second location coupled between each of said multiple upstream channels [channels from the first location to the second location] (501 and 502) and said downstream channel [channel from said second location to said first location] (516) (convolving each of said estimated impulse responses with a signal on the corresponding one of said [channels from the first location to the second location] to generate an estimate corresponding [to] each of said [channels from the first location to the second location]; and summing the individual estimates, according to the alternate meaning applied to the term "impulse response" in claims 26 and 33); and

generating an error signal (the output of subtracter 539) representing the difference between a desired signal on said downstream channel and a sum of said estimates (535 and 536) (inherently comprising summing each of said individual estimates) and transmitting said error signal (516) to said first location.

Hirano does not disclose that said estimate[s] [are] generated using a frequency domain recursive least squares algorithm in the prior-art system of Fig. 2; however, Hirano disclose at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter of the invention may operate in the frequency domain. At the time the present invention was made, the RLS algorithm was well known in the art to provide superior performance (faster convergence and better tracking) relative to the more commonly used LMS (least mean squares) algorithm in applications where the undesired signal is highly correlated with the desired signal (a condition recognized by Hirano et al. for the multi-channel echo canceling arrangement - column 4, lines 6-11). (Since adaptive filter 531 in the prior-art arrangement should ideally only cancel the portion of the mixed signal 514 due to the echo of received signal 501, while allowing adaptive filter 532 to cancel the portion of mixed signal 514 due to the echo of received signal 502, as was well known in the art, the received signal 502 of the other channel is a "desired" signal with respect to the operation of adaptive filter 531.) Also, frequency-domain adaptive filtering techniques were known to provided more efficient implementations for adaptive filters having a large number of taps (as typically

required for echo cancellation). And Nam discloses a frequency domain block RLS adaptive algorithm that is useful for identification of nonlinear system that using adaptive filtering process and for nonlinear distortion in communication system (i.e., echo system). And the frequency domain block RLS adaptive algorithm has the advantage as a power technique in transfer function approach for obtaining statistically meaningful results using the frequency domain block RLS adaptive algorithm (Section 4, Conclusion on page 2409-2410 of the IEEE Paper).

Therefore, it would have been obvious to one of ordinary skill in the art to modify the multi-channel echo canceling apparatus of Hirano with a more powerful frequency domain block RLS adaptive algorithm as taught by Nam in adaptive filtering transfer function processing in order to obtaining statistically meaningful results.

4. **Claims 2-3, 7-8, 12-13, 19-20, 25, 27-28, and 34-35** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano and Nam as discussed above, further in view of Mansour and Gray ("Unconstrained Frequency-Domain Adaptive Filter" [Reference V]).

Regarding **claims 2-3, 7-8 and 12-13**, as described above, Hirano and Nam discloses an apparatus and associated method of normal operation meeting the limitations of claims 1, 6, and 11. Hirano and Nam do not disclose that the adaptive filter of the apparatus generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix formed by augmentation of said input signal (or as more specifically claimed in claims 3, 8, and 13, forming a matrix of vectors representing said input signal; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

Mansour and Gray disclose generally an adaptive filter for use in applications such as echo cancellation (page 726, first paragraph) that generates an estimate of an impulse response in part by diagonally decomposing by Fourier transformation a circulant matrix ($\mathbf{X}_k$) formed by augmentation of an input signal. Mansour and Gray do not describe the formation of the circulant matrix $\mathbf{X}_k$ as comprising the separate steps of forming a matrix of vectors representing said input signal and augmenting the matrix to form a circulant matrix; rather the document implies a more direct formation of the circulant matrix $\mathbf{X}_k$ by forming a vector of length $2N$ of consecutive input samples and creating a circulant matrix by placing that vector in the first row, then

forming each consecutive row by rotating the row above to the right one position. The matrix $\mathbf{X}_k$ in Equation 7 of page 727 of Mansour and Gray can be resolved (by separating it into four N x N matrices) into an N x N matrix **X** (occurring twice in the augmented matrix) and a matrix **X'** (also occurring twice in the augmented matrix) equivalent to that described by Applicant at page 20 of the specification as follows:

$$\chi_k = \mathbf{C} = \begin{bmatrix} \mathbf{X'} & \mathbf{X} \\ \mathbf{X} & \mathbf{X'} \end{bmatrix}, where$$

$$\mathbf{X} = \begin{bmatrix} x(N) & x(N+1) & x(N+2) & \dots & x(2N-1) \\ x(N-1) & x(N) & x(N+1) & \dots & x(2N-2) \\ x(N-2) & x(N-1) & x(N) & \dots & \dots \\ \dots & \dots & \dots & \dots & x(N+1) \\ x(1) & x(2) & \dots & x(N-1) & x(N) \end{bmatrix} and$$

$$\mathbf{X'} = \begin{bmatrix} x(0) & x(1) & x(3) & \dots & x(N-1) \\ x(2N-1) & x(0) & x(1) & \dots & x(N-2) \\ x(2N-2) & x(2N-3) & x(0) & \dots & \dots \\ \dots & \dots & \dots & \dots & x(1) \\ x(N+1) & x(N+2) & \dots & x(2N-1) & x(0) \end{bmatrix}$$

Thus, matrix $\mathbf{X}_k$ of Mansour and Gray is equivalent to that claimed by Applicants; and Applicants have not shown any benefit to forming such a matrix by the separate steps claimed. Matrix ($\mathbf{X}_k$) is then diagonally decomposed by Fourier transformation to

form the diagonal matrix $\mathbf{X}_k$ (which could be named "$\boldsymbol{D}$" without the exercise of any inventive process). Mansour and Gray also do not describe the frequency-domain adaptive filter as a "recursive least squares" filter; however, since the complex conjugate transpose (a.k.a. the Hermitian) of a diagonal matrix (i.e., "$\boldsymbol{D}$") is equivalent to the complex conjugate of the matrix, Applicants' Equations 15 and 16 at page 25 of the specification are equivalent to the equations of claim 5, which claim must include all the limitations of claim 1, and therefore must define a frequency domain recursive least squares algorithm, Applicants admit at page 26, lines 2-3 of the specification with regard to Equations 15 and 16, that "This algorithm is exactly the unconstrained frequency-domain adaptive filter proposed by Mansour and Gray". Thus, to the same extent that the adaptive filter of Applicants' invention employs a frequency-domain recursive least squares algorithm, so does that of Mansour and Gray. Mansour and Gray disclose in the abstract on page 726 that for a large number of taps (as required in typical acoustic echo canceling applications) the disclosed adaptive filter offers significant reduction in computational requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano et al. at column 4, lines 6-10).

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ the frequency-domain adaptive filter of Mansour and Gray in the multi-channel echo canceling method and apparatus of Hirano and Nam by providing circuits to perform each of the recited steps in order to obtain the benefits described in the abstract of Mansour and Gray.

5. Regarding **claims 19-20, 27-28, and 34-35**, Hirano and Nam do not disclose that the adaptive filter of the prior art apparatus and method of generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix formed by augmentation of said input signal (or as more specifically claimed in claims 20, 28, and 35, by forming a matrix of vectors representing said input signal [on said upstream channel]; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

As described above regarding claims 2, 3, 7, 8, 12, and 13, Mansour and Gray disclose a frequency-domain adaptive filter for applications such as acoustic echo cancellation that for a large number of taps (as required in typical acoustic echo canceling applications) offers significant reduction in computational

requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano at column 4, lines 6-10). As further described above in regard to claims 2, 3, 7, 8, 12, and 13, the adaptive filter and method of Mansour and Gray is equivalent to that claimed, and the claimed adaptive filter and method are an obvious variation of that of Mansour and Gray.

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ the frequency-domain adaptive filter of Mansour and Gray in the multi-channel echo canceling method and apparatus of the prior art disclosed by Hirano and Nam by providing circuits to perform each of the recited steps in order to obtain the benefits described in the abstract of Mansour and Gray.

Regarding **claim 25**, in the apparatus and associated inherent method of operation of the prior-art echo canceller of Fig. 2 of Hirano, employed in a teleconferencing system as described at column 1, lines 18-36, the multiple channels of upstream information (501 and 502) comprise sound generated at the first location and the distortion paths comprise echo paths (505 and 506) at the second location coupled between each of said multiple upstream channels and said downstream channel (516).

6. **Claims 38 and 46** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano and Nam, and further in view of Benesty et al. ("A Better Understanding and an Improved Solution to the Problem of Stereophonic Acoustic Echo Cancellation" [Reference U]).

Regarding **claims 38 and 46**, Hirano discloses in Fig. 2 a prior-art multi-channel teleconferencing apparatus comprising:

at least first and second upstream electrical paths (501 and 502) between a first location (not illustrated, but inherently present in a teleconferencing system as disclosed at column 4, lines 1-6) and a second location (as illustrated in Fig. 2) for transmitting acoustic signals from said first location to said second location;

at least one downstream electrical path (516) between said second location and said first location for transmitting acoustic signals from said second location to said first location;

a finite impulse response filter (the combination of 531 and 532) coupled between said upstream paths (501 and 502) and said downstream path (516) for generating an estimate of an impulse response corresponding to echo paths (505 and 506) at said second location coupled between said at least first and second upstream channels and said downstream channel; and

a difference circuit (105) for generating an error signal (26) representing the difference between a signal (24) on said downstream channel representing sound at said second location and said estimate (the output of adaptive filter 103).

Hirano do not disclose at least one non-linear transformation module coupled within each of one or more of said upstream paths, nor that the estimate is generated in the prior-art echo canceller of Fig. 2 using a frequency domain recursive least squares algorithm.

Hirano disclose at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter of the invention may operate in the frequency domain. At the time the present invention was made, the RLS algorithm was well known in the art to provide superior performance (faster convergence and better tracking) relative to the more commonly used LMS (least mean squares) algorithm in applications where the undesired signal is highly correlated with the desired signal (a condition recognized by Hirano et al. for the multi-channel echo canceling arrangement - column 4, lines 6-11). (Since adaptive filter 531 in the prior-art arrangement should ideally only cancel the portion of the mixed signal 514 due to the echo of received signal 501, while allowing adaptive filter 532 to cancel the portion of mixed

signal 514 due to the echo of received signal 502, as was well known in the art, the received signal 502 of the other channel is a "desired" signal with respect to the operation of adaptive filter 531.) Also, frequency-domain adaptive filtering techniques were known to provided more efficient implementations for adaptive filters having a large number of taps (as typically required for echo cancellation). And Nam discloses a frequency domain block RLS adaptive algorithm that is useful for identification of nonlinear system that using adaptive filtering process and for nonlinear distortion in communication system (i.e., echo system). And the frequency domain block RLS adaptive algorithm has the advantage as a power technique in transfer function approach for obtaining statistically meaningful results using the frequency domain block RLS adaptive algorithm (Section 4, Conclusion on page 2409-2410 of the IEEE Paper).

Therefore, it would have been obvious to one of ordinary skill in the art to modify the multi-channel echo canceling apparatus of Hirano with a more powerful frequency domain block RLS adaptive algorithm as taught by Nam in adaptive filtering transfer function processing in order to obtaining statistically meaningful results.

Benesty et al. (including Applicants) disclose in Reference U a method of improved stereophonic echo cancellation in which the problem of a high degree of correlation between the signals of the two "upstream" channels is partially addressed by placing a "non-linear transformation module" in each upstream signal path (page 305). Benesty et al. disclose at pages 305-306, sections 6 and 7 that the non-linear transformation improves the operation (reduces the degree of misalignment) of the stereophonic echo canceller.

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to further to employ the non-linear transformation module of Benesty et al. in order to reduce the degree of correlation between the "upstream" channels and thus further improve the level of performance of the echo canceller as taught by Hirano and Nam.

7. **Claims 39-40, and 47-48** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano, Nam and Benesty et al as applied to claims 38 and 46 above, and further in view of Mansour and Gray (("Unconstrained Frequency-Domain Adaptive Filter" [Reference V]).

Regarding **claims 39, 40, 47, and 48**, Hirano, Nam and Benesty et al do not disclose that the adaptive filter of the prior art apparatus and method of Fig. 2 generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix formed by augmentation of said input signal (or as more specifically claimed in claims 40 and 48, by forming a matrix of vectors representing said input signal [on said upstream channel]; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

As described above regarding claims 2, 3, 7, 8, 12, and 13, Mansour and Gray disclose a frequency-domain adaptive filter for applications such as acoustic echo cancellation that for a large number of taps (as required in typical acoustic echo canceling applications) offers significant reduction in computational requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano at column 4, lines 6-10). As further described above in regard to claims 2, 3, 7, 8, 12, and 13, the adaptive filter and method of Mansour and Gray is equivalent to that claimed, and the claimed adaptive filter is an obvious variation of that of Mansour and Gray.

Allowable Subject Matter

8. Claims 4-5, 9-10, 14-17, 21-24, 29-31, 36-37, 41-45, and 49-52 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Response to Arguments

9. Applicant's arguments with respect to claims 1-31, 33-52 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

10. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Chen et al, Agazzi, and Chu are made of record here as pertinent art to the claimed invention. The cited references disclose Volterra signal processing (as discussed in Nam) is being used for various filtering processes in echo canceller.

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Xu Mei whose telephone number is 571-272-7523. The examiner can normally be reached on Monday-Friday (9:30-6:00).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Vivian Chin can be reached on 571-272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

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Xu Mei
Primary Examiner
Art Unit 2615
09/28/2006